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Apparatus for Measurement of Non-Linear Distortion as a Continuous Function of Frequency

by

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(Research Department, BBC Engineering Division)

BRITISH BROADCASTING CORPORATION

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FOREWORD

THIS is one of a series of Engineering Monographs published by the British Broadcasting Corporation.

About six are produced every year, each dealing with a technical subject within the field of television and sound broadcasting. Each Monograph describes work that has been done by the Engineering Division of the BBC and includes, where appropriate, a survey of earlier work on the same subject. From time to time the series may include selected reprints of articles by BBC authors that have appeared in technical journals. Papers dealing with general engineering developments in broadcasting may also be included occasionally.

This series should be of interest and value to engineers engaged in the fields of broadcasting and of telecommunications generally.

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APPARATUS FOR THE MEASUREMENT OF NON-LINEAR DISTORTION AS A CONTINUOUS FUNCTION OF FREQUENCY

SUMMARY

Measurements of the non-linear distortion products generated by a system, such as a loudspeaker, which has more than one degree of freedom must be made as a continuous function of frequency if a true assessment of the distortion is to be obtained. Various methods of performing such measurements are reviewed and the limitations are indicated. A description is then given of a new system by which the individual harmonic and intermodulation distortion products can be measured as a continuous function of frequency. In addition the working principles are described of a new type of stroboscope which is capable of operating at constant illumination up to unusually high repetition rates; this device is suitable for examining vibrating systems for resonance modes which may give rise to distortion through excessive amplitude.

Details are given of the circuits employed and the performance obtained from an equipment constructed on these principles, and the results of some measurements are shown.

1. General

Most parts of a broadcasting chain generate some non-linear distortion which will require investigation. The degree of distortion produced by equipment such as amplifiers will vary only slowly with frequency and can therefore be sufficiently defined by measurements at a few selected frequencies. On the other hand, certain types of apparatus possess more than one degree of freedom and exhibit non-linear distortion effects which vary rapidly with frequency; for example, a high-quality loudspeaker having an axial response curve in which the irregularities are no greater than 3 dB may produce non-linear distortion which varies irregularly, even over a narrow frequency range, by more than 20 dB. For measurements of distortion to be meaningful in these circumstances it is clearly necessary to measure the distortion products as a continuous function of frequency. It is further desirable to measure individually the various orders of distortion product, such as 2nd harmonic, 3rd harmonic, and so on, not only for the purposes of analysis and design, but also because the common practice of taking the r.m.s. total of these products as a subjective index of distortion has proved to be misleading in cases where the series converges only slowly.^{1,2}

Existing methods employing pure tones to measure non-linear distortion as a continuous function of frequency are of three kinds. In the first, a gliding-tone signal is applied and the harmonics in the output are separated from the fundamental by a series of filters with different cut-off frequencies, appropriate switching operations being carried out mechanically as the measurement proceeds.^{3,4} The degree of resolution obtained by the filters is usually limited by considerations of instrumental complication and the result commonly includes several harmonics together with a certain amount of unwanted noise. In the second type of test, a low-frequency and a high-frequency tone are simultaneously applied,⁵ and the coefficient of modulation of the higher-frequency signal by the lower-frequency signal is determined by conventional demodulating circuits. In this case, the higher frequency may be

fixed and the lower variable within the range determined by the side bands produced; the test is in practice restricted to the evaluation of the r.m.s. sum of the first few side-band components. As in the case of harmonic distortion measurements, however, it would be preferable to isolate the individual distortion products. The third method⁶ also employs two tones, in this case separated by only a small frequency difference; the frequencies of the two tones can both be varied continuously, keeping the difference constant throughout the measurement. Some types of distortion product with fixed frequencies which are multiples of the original frequency difference can then be isolated by corresponding bandpass filters. There appears to be very little reference in the literature to any measurements other than the simple sum or difference frequency. The essential difference between the second and third types of test is that the second gives emphasis to distortion produced by an odd-order transfer characteristic, i.e. an input/output characteristic containing odd-order powers, whereas the third indicates distortion generated by an even-order characteristic only.

In practice, the usefulness of each of these methods is thus seen to be limited. No harmonics can be measured when the fundamental is less than an octave below the upper frequency cut-off of the system under test. Intermodulation measurements employing two widely spaced test tones are applicable at low frequencies but inconvenient at high frequencies. If the tones are closely spaced, the converse applies; moreover, such a test does not reveal odd-order distortion terms.

Details have also been published⁷ of a further method which may be regarded as similar to the second and third described above, but in which the number of test tones has been increased from two to infinity. The signal employed consists of random noise, the spectrum of which is uniform and continuous except over a narrow frequency range from which the signal is removed by a band-stop filter. The products of non-linear distortion will include both harmonic and intermodulation components lying within this

band; these components are isolated by means of a band-pass filter, the output from which is used as a measure of the distortion. It is not possible, however, to determine the contribution to the total distortion by individual components, and although the centre frequency of the two filters concerned can be varied, the measurement gives no indication over what portion of the spectrum the system is non-linear; a further difficulty arises in assigning to the input signal a peak level which could be related to the rated overload level. Except for its original purpose, which was the testing of carrier communication systems, the method therefore appears to be less useful than those described above.

The apparatus to be described⁸ was designed in 1953 for measurements on loudspeakers and has been in use since 1956. It enables the single- and two-tone tests to be carried out in such a way that harmonic and intermodulation products are measured individually as a continuous function of frequency, the results being substantially independent of noise or other spurious signals in the system under test. The higher levels of distortion are often associated with flexural modes of resonance of the cone or surround. To aid in locating the distorting element, a special stroboscope capable of operating at repetition rates up to 20 kc/s has been included in the equipment.

The apparatus will be described as it would be used to measure the performance of a loudspeaker; the principles employed are given in the following section, details of the actual circuits and their performance are in Section 3, and some typical measurements are given in Section 4.

2. Principles of Design

2.1 Method of Measurement of the Fundamental Component of a Signal

The method employed to measure the level of the fundamental component is not original; it was developed independently in various countries before the war. A description is, however, included both for completeness and because it introduces the novel methods to be described for the measurement of harmonics and odd-order intermodulation products.

Fig. 1 indicates, in simplified form, the method employed to generate a signal and to measure the level of its fundamental component. Signals from a fixed frequency oscillator O_1 and a variable frequency oscillator O_2 , having frequencies f_1 and f_2 respectively ($f_2 \leq f_1$) enter a modulator MOD_1 . The difference frequency $f_1 - f_2$, which lies in the audio-frequency range, is selected by a low-pass filter LPF_1 and applied to the loudspeaker LS , which is being tested. The signal from the measuring microphone MIC , together with a signal of frequency f_2 , derived from O_2 , are applied to a second modulator MOD_2 . From MOD_2 , a signal of the sum frequency $(f_1 - f_2) + f_2$, i.e. f_1 , is selected by the bandpass filter BPF_1 and the amplitude of this signal is recorded by a meter RM . It should be noted that the frequency of the summation product at the output of MOD_2 is independent of the frequency of the signal $f_1 - f_2$ applied to the loudspeaker. By making the pass-band of BPF_1 very narrow, a high degree of discrimination is provided against signals of frequencies other than f_1 leaving MOD_2 and therefore against signals from the microphone having fre-

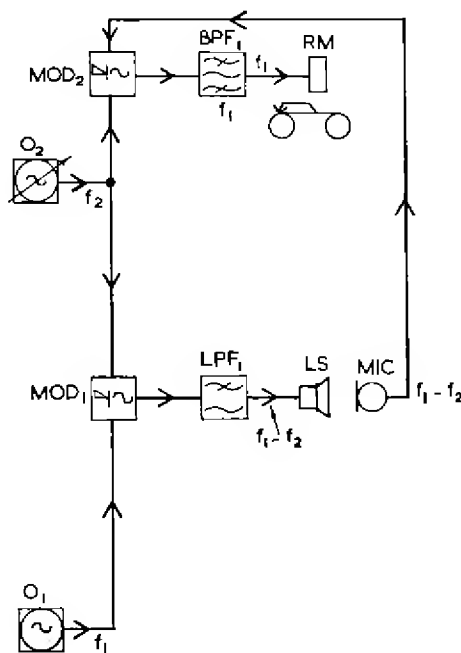


Fig. 1 — Schematic circuit for measurement of fundamental.

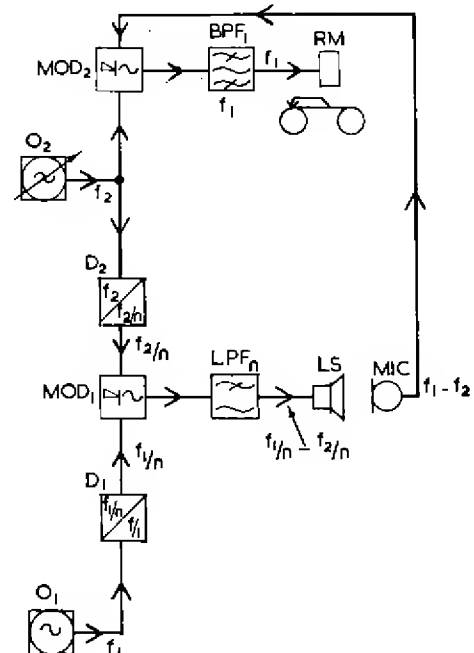


Fig. 2 — Schematic circuit for measurement of harmonics.

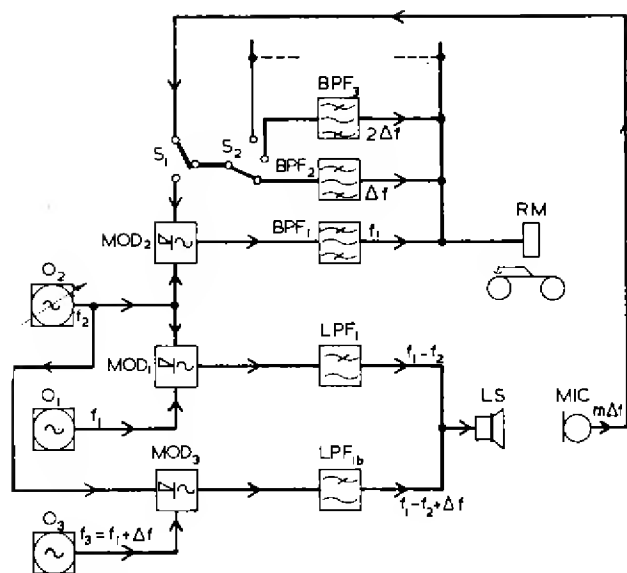


Fig. 3a — Schematic circuit for measurement of even-order intermodulation products.

quencies other than $f_1 - f_2$. As a result, neither distortion products radiated by the loudspeaker nor ambient noise picked up by the microphone affect the level recorded by RM.

2.2 Method of Measurement of Harmonics in a Signal

To measure the amplitude of the n th harmonic generated by the loudspeaker, the circuit of Fig. 1 is extended⁸ as shown in Fig. 2. The frequency f_2 from oscillator O_2 is adjusted so that $f_1 - f_2$ is now n times the frequency of the signal to be applied to the loudspeaker. The outputs from oscillators O_1 and O_2 are taken to separate frequency dividers D_1 and D_2 which yield output signals having frequencies f_1/n and f_2/n respectively. These two signals are applied to the modulator MOD_1 , from which the difference frequency $f_1/n - f_2/n$, which is in the audio-frequency range, is selected by the low-pass filter LPF_n and applied to the loudspeaker. Since, however, the indicating circuit consisting of MOD_2 , BPF_1 , and RM responds only to audio-frequency signals of frequencies in the immediate neighbourhood of $f_1 - f_2$, no output will be registered if the loudspeaker transfer characteristic is linear. If, however, the loudspeaker generates a harmonic of order n , the signal from the microphone will contain a component of frequency $n(f_1/n - f_2/n)$, i.e. $f_1 - f_2$; a corresponding signal of frequency f_1 will then reach RM, which will thus record the amplitude of the n th harmonic only.

2.3 Method of Measurement of Intermodulation Distortion

Fig. 3a shows a further extension of the circuit of Fig. 1 to permit measurement of the intermodulation products generated when two signals, of a constant frequency difference Δf , are simultaneously applied to the system under

test. Signals from O_2 , and from an additional fixed oscillator O_3 of a frequency $f_3 = f_1 + \Delta f$, are taken to an additional modulator MOD_3 . The difference signal from MOD_3 , the frequency of which is $f_3 - f_2$, i.e. $(f_1 - f_2) + \Delta f$, is selected by an additional low-pass filter LPF_{1b} and applied, together with the existing signal of frequency $f_1 - f_2$, to the loudspeaker. If the loudspeaker transfer characteristic is non-linear, the output of the microphone will generally contain, in addition to signals at the original frequencies $(f_1 - f_2)$ and $(f_1 - f_2) - \Delta f$ and harmonics of these, intermodulation products of (among others) frequencies $(m\Delta f)$ and $(f_1 - f_2) - m\Delta f$, where m is an integer.

The components of frequencies $m\Delta f$ are associated with even-order terms in the transfer characteristic of the loudspeaker. Since these frequencies do not vary with f_2 but are constant, they can be isolated by operating the switches S_1 and S_2 so as to introduce one or other of a series of fixed-frequency bandpass filters BPF_2 , BPF_3 corresponding to $m = 1$, $m = 2$, and so on, and the level can then be measured by the meter RM.

Components of frequencies $(f_1 - f_2) - m\Delta f$ are representative of the odd-order terms in the transfer characteristic. Being of variable frequency, they cannot be isolated simply by fixed filters. This difficulty is overcome⁸ by increasing the frequencies of all three oscillators as in Fig. 3b by $m\Delta f$ so that

$$\begin{aligned} O_1 \text{ gives } f_1' &= f_1 + m\Delta f \\ O_2 \text{ gives } f_2' &= f_2 + m\Delta f \\ O_3 \text{ gives } f_3' &= f_3 + m\Delta f = f_1 + (m+1)\Delta f, \end{aligned}$$

the frequencies of the two signals applied to the loudspeaker then remaining as before $(f_1 - f_2)$ and $(f_1 - f_2) + \Delta f$.

Switch S_1 is now operated to connect the output of the microphone to modulator MOD_2 ; the distortion component of frequency $(f_1 - f_2) - m\Delta f$ from MIC together with the frequency of f_2' i.e. $f_2 + m\Delta f$, from oscillator O_2 then yields a sum signal, having the fixed frequency f_1 , which is filtered and registered on RM as before.

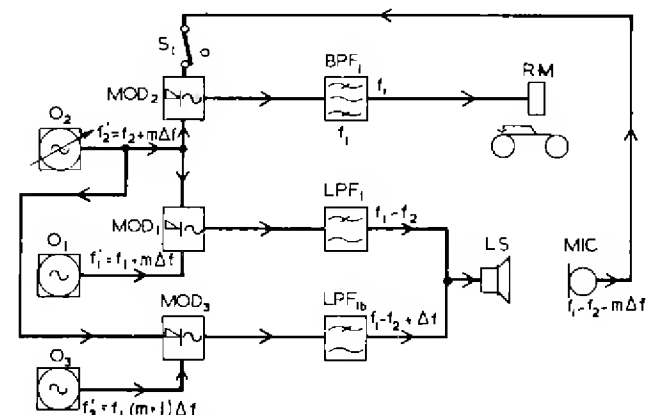


Fig. 3b — Schematic circuit for measurement of odd-order intermodulation products.

2.4 Stroboscope

As described in the previous section two audio-frequency glide tones with a fixed frequency difference Δf are produced by the apparatus; one of them is also employed to control a stroboscopic light source while the other is used to drive the loudspeaker. It has been found that resonance modes of the cone are most easily observed when Δf is about 4 c/s; in fact f_3 has been made variable so that Δf can be adjusted over the range of 0 to 10 c/s.

The type of gas-discharge tube usually employed as a light source for a stroboscope cannot be used at a repetition rate above about 400 c/s because of the length of time required for de-ionization after each discharge. The stroboscope used in these experiments employs a light source consisting of a form of cathode-ray tube, the phosphor having a decay time of the order of 1 microsecond. In order to protect the cathode-ray tube against overload and also to use it more efficiently than is customary with a discharge tube, a novel form of pulsing is employed. The usual method is to apply a very short pulse of constant duration, irrespective of the repetition rate, with the result that at low repetition rates the light output is unnecessarily low. To overcome this defect and give the maximum illumination of the object under test at all frequencies without blurring of the image, the time constant of the pulsing circuit which controls the high-tension current in the tube is automatically adjusted by the frequency of the signal in such a way that the duration of the pulse is substantially a constant fraction of a cycle;⁸ this fraction can have any value between $\frac{1}{4}$ and $\frac{1}{30}$.

One way of producing this pulse is shown diagrammatically in Fig. 4. The input signal waveform is squared by diodes X_1 X_2 , and differentiated by CR to form pulses; a d.c. voltage proportional to the frequency of the input signal is also extracted in the manner shown. The pulses are backed off by this variable d.c. voltage so that the duration of the pulse is maintained at an approximately constant fraction of a cycle for a wide range of frequencies. This residual pulse is then squared and amplified by subsequent valve stages. To increase the total frequency range which can be covered, the value of the time constant CR can be changed.

As an alternative, the time constant of the CR circuit

can be controlled as in Fig. 5 by including one of the elements in the feedback loop of an amplifier whose gain is varied with frequency by the d.c. voltage obtained in the manner previously described.

3. Circuit Details and Performance

3.1 General

Circuit and performance details of an experimental instrument are given in this section. It is not claimed that the circuits used are necessarily the best, and in fact various possible improvements are indicated. Figs. 6 to 11 show the circuits employed and should be referred to in conjunction with the schematic diagrams already described.

3.2 Measurement of the Fundamental and of Harmonics in a Signal

3.2.1 Circuit Details

General

The combination of O_2 , MOD_2 , BPF_1 , and RM , shown in Fig. 1, is to be found in the well-known heterodyne type of wave analyser, and an early commercial instrument of this type was adapted for the purpose; Fig. 6 shows the circuit diagram. The objects of the changes were to obtain an additional output from the oscillator and from the crystal filter, to improve the performance of the filter and the frequency stability of the oscillator and to produce a frequency scale giving a logarithmic law with respect to the audio-frequency $f_1 - f_2$. Such a scale is very desirable as recorded curves of fundamental and of harmonic distortion can then be directly aligned with respect to one another, as in Fig. 7, so that they show the levels of the harmonics for a given fundamental frequency.

The oscillator frequency can be varied from approximately 50 kc/s to 34 kc/s; to reduce the level of the carrier frequency components in the output the modulator, MOD_2 , is of the balanced variety. The bandwidth of the filter BPF_1 , is largely determined by the two quartz crystals and is 4 c/s between the half-power points.

Divider D_2

In Fig. 2 it will be seen that dividers D_1 and D_2 are interposed between oscillators O_1 and O_2 and the modulator MOD_1 ; when the division ratio is unity, i.e. the frequency

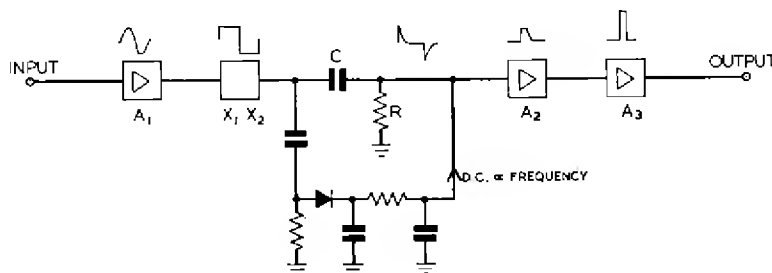


Fig. 4 — Schematic pulsing circuit for high-frequency stroboscope.

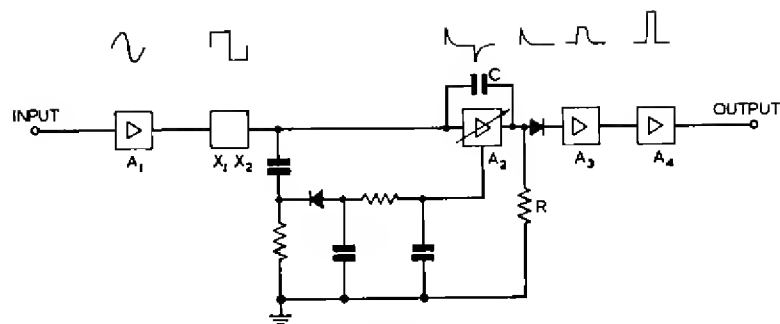


Fig. 5 — Alternative pulsing circuit for high-frequency stroboscope.

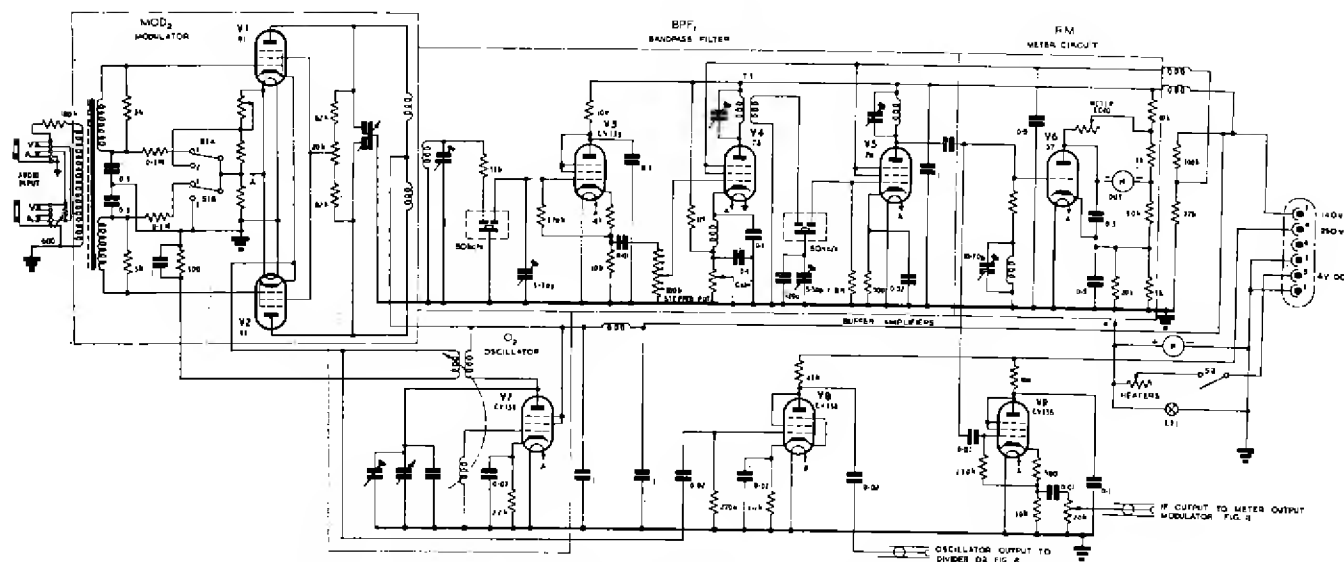


Fig. 6 — Circuit of modified wave analyser (O_2 , MOD_2 , BPF_1 , and RM).

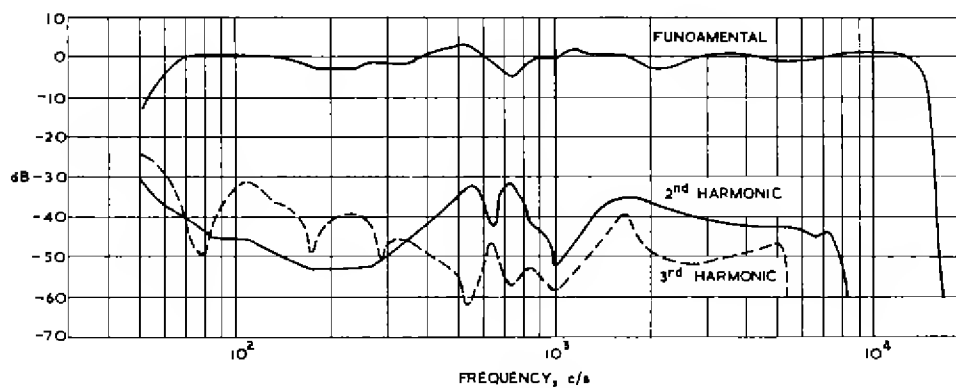


Fig. 7 — Method of showing levels of fundamental, second, and third harmonics for a given fundamental frequency.

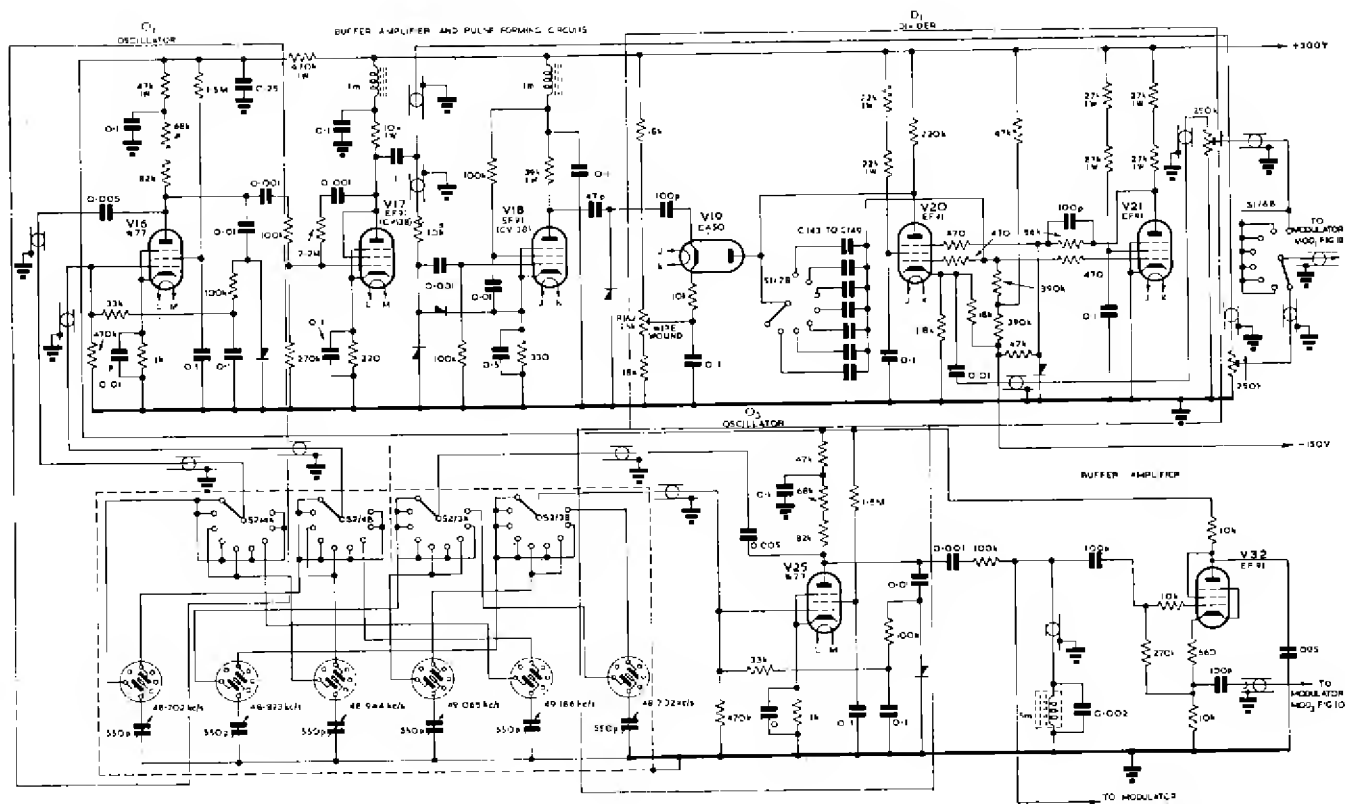


Fig. 9 — Circuit of oscillators O_1 and O_2 and Sanaphant divider D_1 .

of the output signal is the same as that of the input, the circuit is effectively the same as that of Fig. 1. The signal applied to divider D_2 may vary between 50 kc/s and 34 kc/s; a divider which will operate only with an input of fixed frequency is therefore unsuitable. Fig. 8 shows the circuit used;⁹ it is of the binary type and was chosen as experience had shown it to be extremely stable and reliable. Three binary pairs are used, giving division ratios down to 1 : 8, thus allowing measurements to be made of harmonics up to the 8th. Two binary pairs would have restricted the analysis to the 4th harmonic; on the other hand, the additional instrumental complication needed to make use of an additional binary pair to go beyond the 8th was thought to be excessive compared with the usefulness of the information gained. The sine waveform input from the oscillator O_2 is squared by rectifiers MR_1 and MR_2 . In the 'unity' position of the division ratio selector switch, this square wave signal is connected to the output switch; for other positions it is differentiated by $C5R6$ to form a pulse. Frequency-division of the repetition rate of this pulse is provided by one or more of the binary pairs $V2V3$, $V4V5$, and $V6V7$ in the usual way, giving division ratios of 1 : 2, 1 : 4, and 1 : 8. From the output of the last pair, a pulse delayed by $L1$ $C29$ is returned through $V11$, $V12$, $V13$, and/or $V14$ to the appropriate binary pairs to produce the

ratios 1 : 3, 1 : 5, 1 : 6, and 1 : 7. Further details of this circuit are given in the reference cited.

Oscillator O_1

It was shown in Section 3.1 that the frequency f_1 which is intended to be passed by the filter BPF_1 is derived from oscillator O_1 . (See Fig. 2.) In order to obtain the desired selectivity, the filter has a uniform response over a band of only two cycles per second, and it follows that the frequency drift of O_1 must be appreciably less than this; a crystal controlled oscillator with amplitude stabilization is therefore employed. Fig. 9 shows the circuit, together with a buffer amplifier $V17$ to divider D_1 .

Divider D_1

As previously mentioned for divider D_2 the input signal by-passes the divider when unity division ratio is required but the waveform is squared and differentiated for the other ratios. However, unlike the input to D_2 , the signal applied to D_1 has a fixed frequency and the divider may therefore be simpler. The Sanaphant¹⁰ circuit chosen employs only a quarter of the number of valves used for D_2 and is basically of the Miller time-base variety. The division ratio is determined by one of the capacitors $C143$ to $C149$

shown in Fig. 9 and also by the 'catching' voltage which is applied through V19 from the potentiometer R162. It is found that if this voltage is kept constant and the capacitors are used as the element determining the division ratio, the output level varies much less with this ratio than if the alternative arrangement was used.*

Modulator MOD₁

The modulation system used for MOD₁, shown in Fig. 10, was developed by Mayo,¹¹ and was chosen with the object of keeping distortion in the output to a minimum. Currents from high-impedance sources are applied to a pair of rectifiers; the waveform of the switching current is rectangular instead of the customary large amplitude sine wave, whilst the modulated current has a pure sine waveform. Neither of the outputs from the dividers is in the form of a sine wave and one or other of the two signals must therefore be filtered before it can be applied to the modulator. The choice as to which signal is filtered is determined by the fact that the level of any particular harmonic in the output can be reduced by employing a switching waveform which does not contain that harmonic

* This form of divider could probably replace that employed for D₂. In this case the catching voltage would also be varied with the input frequency by mounting the potentiometer R162 on the spindle of the variable capacitor of oscillator O₂.

instead of always using a square waveform. For example, if the 5th harmonic is to be reduced to a minimum a ratio of 1 : 4 is used for the switching waveform, and it is an incidental property of the binary divider employed that this condition is automatically achieved for almost all of the ratios required. In addition it is easier to obtain the equally essential pure sine wave from the fixed-frequency oscillator and divider D₁, than from the variable frequency supplied by D₂. The resonant circuit employed to extract this sine wave from the output of D₁ is shown in Fig. 10. The tuning capacitors C166 to C173 are switched in accordance with the division ratio, whilst the resistors R195 to R202 are used to equalize the output level for each ratio. The lower side-band in the modulator output is selected by the low-pass filter LPF₁₋₈ lying between T1 and T2, the filter cut-off frequency also being switched according to the ratio of the dividers; it may be noted here that if a balanced modulator were employed the carrier level in its output would be considerably reduced and fewer components would be needed in the filter. Negative feedback around the output stage V15, T3, ensures that it introduces very little distortion.

Meter RM

The signal from the crystal filter which is read on the meter RM has a frequency of about 50 kc/s. Very few re-

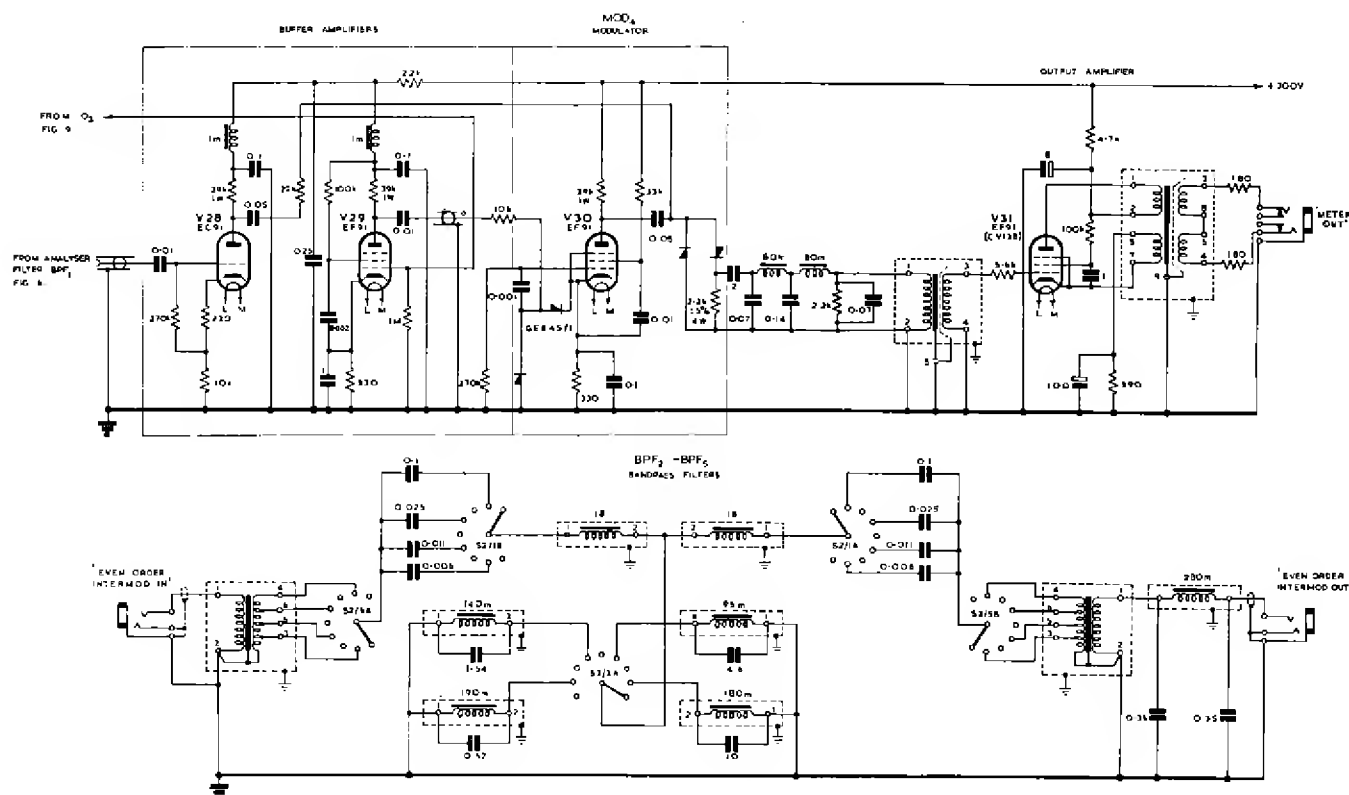


Fig. 11 — Circuit of modulator MOD₁ with associated filter and of band pass filters BPF₁ to BPF₆.

cording instruments will operate with an input of this frequency and therefore to permit continuous recordings to be made an output from the filter is shifted in frequency to a few hundred cycles per second. For this purpose an additional output taken from the filter BPF₁ and a signal from the crystal oscillator O₃ are applied as shown in Fig. 11 to modulator MOD₄ the design of which is similar to MOD₁.

The difference frequency from MOD₄ is 120 c/s, 360 c/s, or 480 c/s according to the particular crystal in use at the time; the low-pass filter following MOD₄ can therefore be very simple.

3.2.2 Performance

3.2.2.1 Measurement of Fundamental

The minimum measurable ratio of fundamental to unwanted signals is determined by the selectivity of the crystal filter BPF₁. The frequency characteristic of the modified filter is flat to within 0.25 dB over a bandwidth of 2 c/s and the half-power points are 4 c/s apart; the attenuation is greater than 20 dB at 8 c/s from the centre of the pass-band and 60 dB at 60 c/s from the centre.

It was shown in Section 3.2 that harmonics present in the loudspeaker output do not affect the measurement of the fundamental and the purity of the input to the loudspeaker is therefore not important for this measurement. In fact at 1 kc/s the 2nd harmonic in the signal supplied is 57 dB, the 4th is 70 dB, and the remainder 80 dB or more below the level of the fundamental.

3.2.2.2 Measurement of Harmonics

The minimum percentage distortion which can be measured for a particular harmonic is determined by the selectivity of the filter BPF₁, by the level of the harmonic produced by the analyser itself and by the noise level. The selectivity is given in Section 3.2.2.1 and is a limitation only at low frequencies, where harmonics may be close to each other or to mains-frequency interference.

Distortion is generated in the instrument by modulator MOD₂ and also by MOD₁ and the output amplifier. That generated by MOD₂ is sufficiently low to permit a harmonic to be measured 80 dB below the fundamental. Harmonics generated in MOD₁ and the output amplifier affect the test signal supplied and the distortion allowable in this signal depends on the measurement being made; for example, if 3rd harmonic is being measured the signal should be free of this component, whereas others such as 2nd, 4th, etc. are not recorded and are therefore relatively unimportant.

The level of the harmonic which is present in the test signal is given below for the different measurement conditions; most of the distortion is generated in the output amplifier and could be reduced if it were required.* However considerably more distortion than this is generated in even the highest quality loudspeakers when operating at normal listening levels.

* The levels of at least the 2nd and the 4th harmonic could be reduced by employing a symmetrical stage instead of a single ended one (V15, Fig. 10).

Measurement of	Level of harmonic with respect to fundamental in test signal of 2V. at	
	50 c/s	1 kc/s
2nd harmonic	-58 dB	-68 dB
3rd harmonic	-56 dB	-80 dB
4th harmonic	-70 dB	< -80 dB
5th harmonic	-76 dB	< -80 dB
6th harmonic	-80 dB	< -80 dB
7th harmonic	< -80 dB	< -80 dB
8th harmonic	< -80 dB	< -80 dB

3.3 Measurement of Intermodulation Distortion

3.3.1 Circuit Details

Most of the items shown in Fig. 3 have already been described. Of the remainder the circuit of oscillator O₃ is the same as that of O₁ and is also shown in Fig. 9. In Fig. 10, modulator MOD₃, similar to MOD₁, is connected directly to the filter LPF_{1b}, and thence to the output stage. The signal from LPF₁ is then added in a linear non-mutual network and the composite signal applied to the loudspeaker.

The frequency difference Δf between the two test signals has been set at 121 c/s, this choice being a compromise. On the one hand a large frequency difference eases the design of the filters necessary to separate the different intermodulation components; on the other hand rapid fluctuations of distortion with frequency can be more accurately resolved if the two signals are confined to a narrow bandwidth. It is also important to avoid interference from harmonics of the mains frequency when measuring the difference products. The geometrical mean between 100 and 150 c/s was therefore chosen.

The circuits employed for the bandpass filters BPF₂, BPF₃, BPF₄, and BPF₅, are given in Fig. 11.

3.3.2 Performance

As mentioned in Section 3.2.2.2 the lowest level of distortion which can be measured is set by the frequency separation of the signals concerned, and by the selectivity of the filters. In the measurement of odd-order distortion terms the selectivity of filter BPF₁ is such that for signals 121 c/s away from the centre of the pass-band frequency the attenuation is about 75 dB. When measuring the even-order terms the fundamental frequencies are much farther removed from the intermodulation products than are the odd-order ones and therefore the filters BPF₂ to BPF₅ need not be so selective to achieve the same rejection of the test signals. Against neighbouring intermodulation components the filters have a discrimination varying from 65 to 45 dB; the attenuation of the comparatively low level 100 c/s and 150 c/s harmonics of mains frequency which might be picked up by the microphone is about 25 dB by BPF₂, 50 dB by BPF₃, and over 70 dB by BPF₄ and BPF₅.

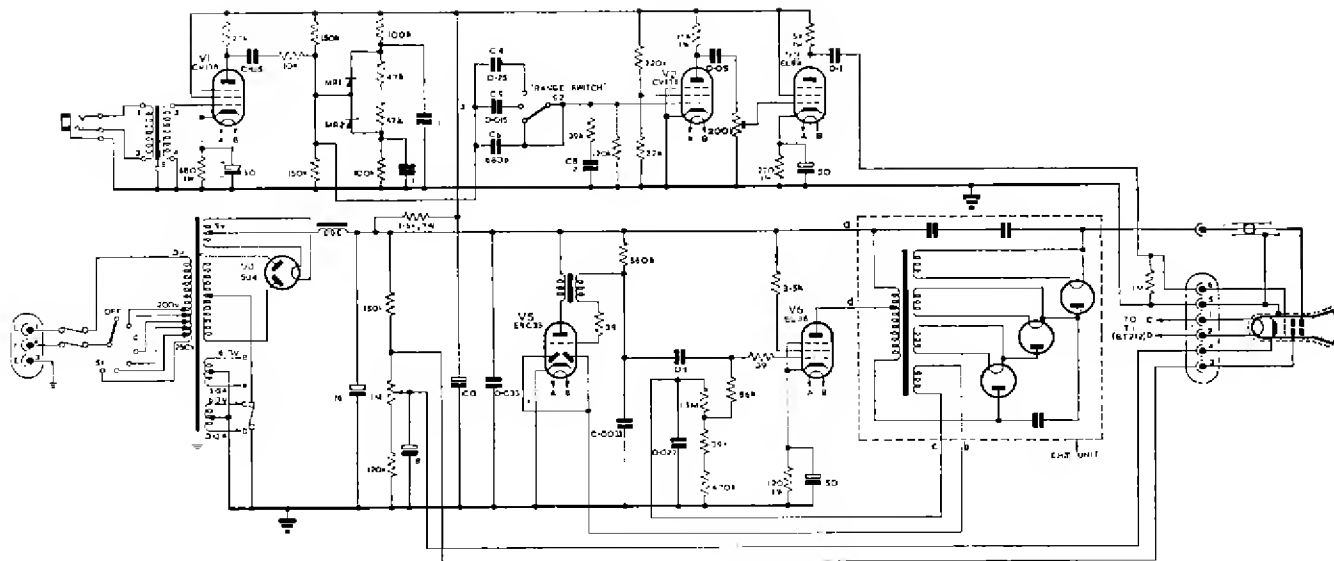


Fig. 12 — Circuit of stroboscope.

3.4 Stroboscope

An embodiment of the scheme of Fig. 4 is shown in Fig. 12. The input signal is amplified by V1, the waveform squared by MR1 and MR2 and then differentiated. Capacitor C4 is used for the frequency range 50–500 c/s, C5 for 400–5,000 c/s, and C6 for 4,000–20,000 c/s. A rectified voltage is provided by the grid of V2 and smoothed by C8; the pulse waveform above this backing off voltage is amplified and also limited by V2. It is further amplified by V3 and applied to the control grid of a special cathode-ray tube. This tube was specially made for the purpose* and is run under rather unusual conditions in order to obtain the maximum light output. The beam is defocused to give an image of approximately 1 in (2.5 cm) diameter, and the dissipation at the screen is about 5 watts mean; at these levels a 'non-solarizing' glass is essential to ensure that none of the light is attenuated. The tube is normally biased off and a peak beam current of approximately 3 mA at 12 kV is drawn when the input pulse is applied. The frequency difference between the signal applied to the loudspeaker and that applied to the tube may be varied between zero and 10 c/s; a figure of about 4 c/s is found to be optimum, and under these conditions it is found that a sharp visual image is obtained using an on/off ratio as high as 1 : 6 which means that a correspondingly high light output is produced by the tube. The phosphor used has a decay time of about 1 μ sec.† In fact the frequency limitation in stroboscopic examination no longer lies in the light source but in obtaining sufficient amplitude of motion of the test object.

* Suitable tubes are now available commercially.

† This permits modes of vibration of microphone ribbons to be examined at frequencies in excess of 10 kc/s.

4. Results

Fig. 13 shows, as an example of the practical utility of the equipment, some results of low-level tests on a small loudspeaker. The effect of employing a vented cabinet on the overall level of 3rd harmonic distortion at low frequencies is clearly evident and in accordance with expectations. More arresting is the fact that at the vent resonance frequency the output from the loudspeaker cone surround shows a considerable increase in the level of distortion in relation to the level of the fundamental. The reason for this effect is that at this frequency a given displacement of the loudspeaker cone generates a much higher back pressure in the cabinet than at neighbouring frequencies. This high pressure displaces the cone surround, in the opposite direction to that taken by the cone, until it reaches the limits of its travel. The same effect, although over a wider frequency band, will be produced by any unit having a very flexible surround and mounted in a small closed cabinet. Similar types of curve taken during the development of high-grade monitoring loudspeakers have proved extremely valuable in locating and eliminating various sources of distortion.

Figs. 14a and b show a set of measurements made on a medium-size monitoring-type loudspeaker in which the crossover frequency between low- and high-frequency units is 1.8 kc/s; the input voltage was held constant at a level which gave a sound pressure of 94 dB w.r.t. 0.0002 dyn/cm² at 4 ft 6 in. at 400 c/s. It will be seen that the distortion level increases as the test frequency approaches the lower end of the band over which the different units operate; although the distortion levels are low it is evident, particularly from Fig. 14b, that the crossover frequency could not have been appreciably reduced without serious

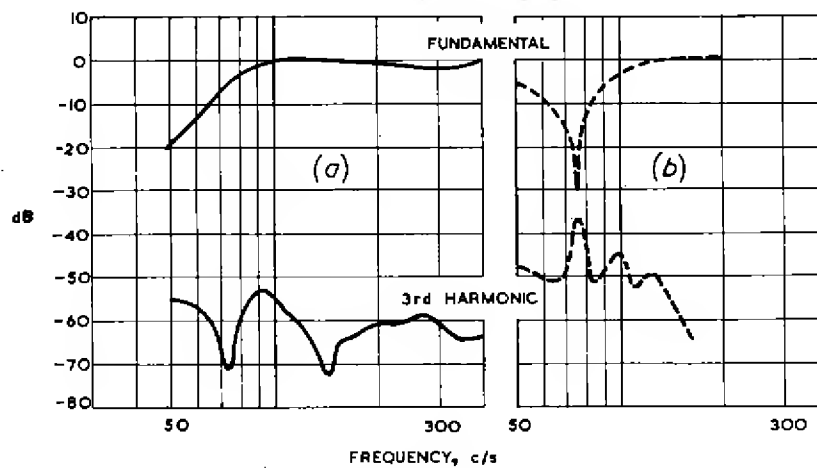


Fig. 13 — Level of third harmonic from small loudspeaker.
 (a) Axial response at 3 ft 6 in. (vent resonance 75 c/s).
 (b) Output from cone surround only (arbitrary zero).

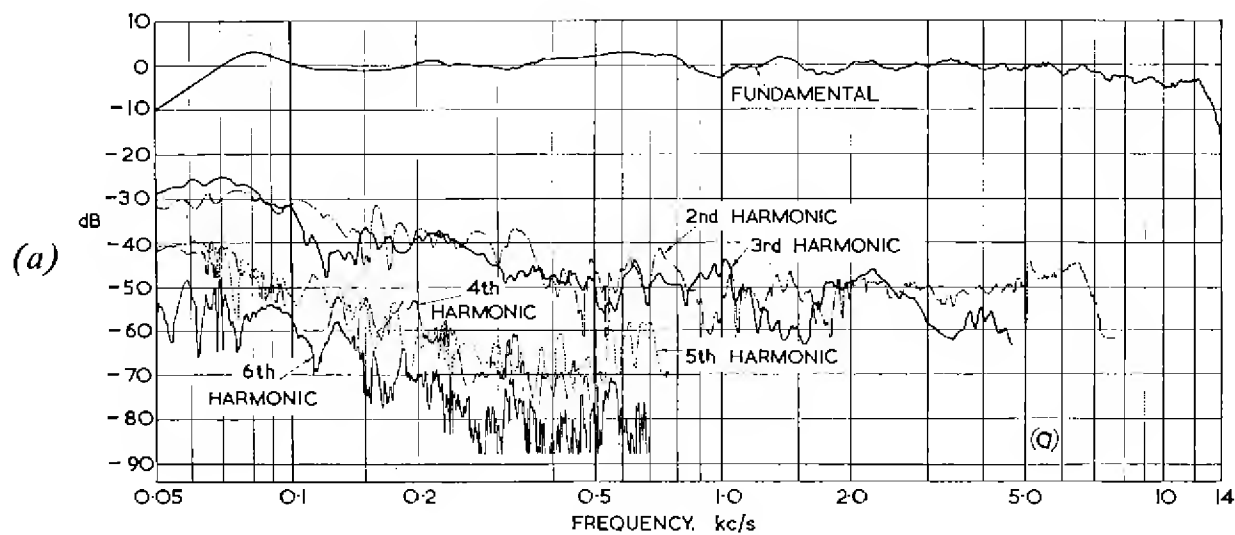
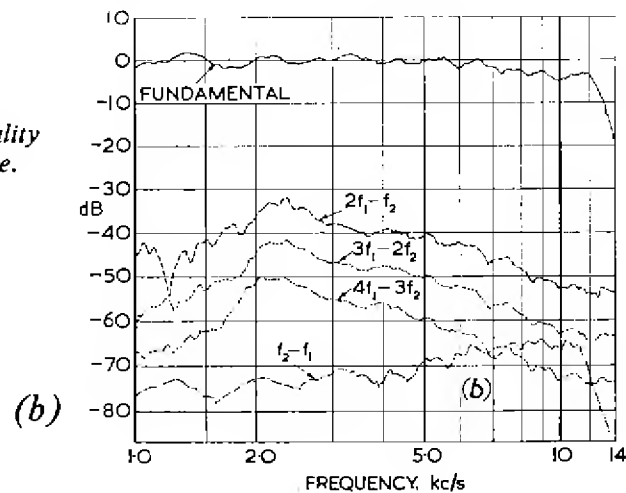


Fig. 14 — Levels of distortion from high-quality two-unit loudspeaker in an unvented enclosure.
 (a) Level of harmonics.
 (b) Level of intermodulation products.



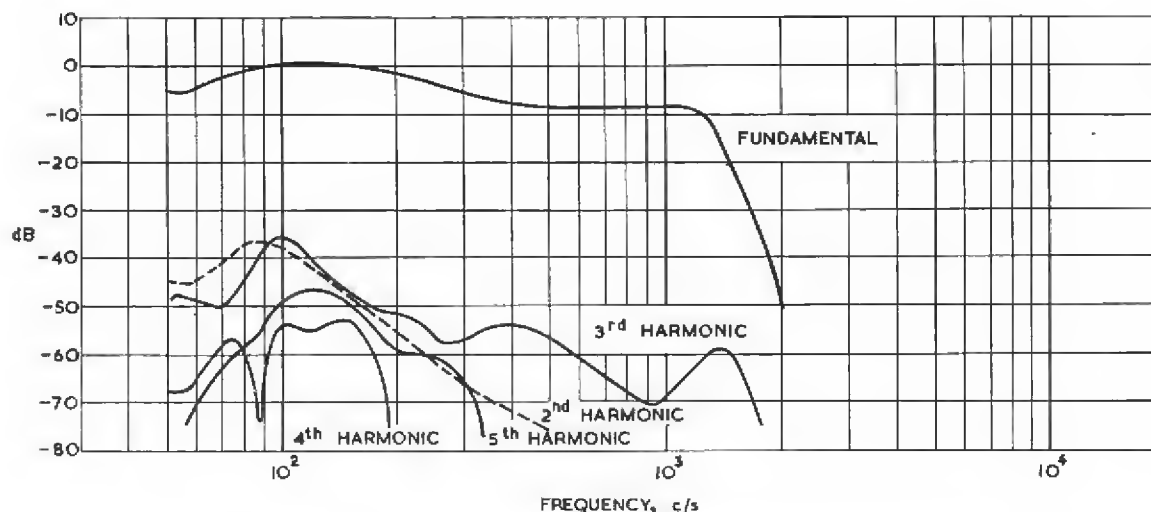


Fig. 15 — Level of harmonics from experimental loudspeaker crossover network.

risk of overloading the high-frequency units. The difference in the magnitude of the irregularities in the curves for fundamental and harmonics should be noted; also the fact that the frequency at which the level of one harmonic is a maximum often does not coincide with that for the maximum of another and may, indeed, even coincide with a minimum, e.g. 3rd and 5th harmonics at 650 and 700 c/s. Measurements of distortion as a continuous function of frequency as opposed to measurements at a few spot frequencies should need no further justification even if the method were only used to choose the frequencies at which to carry out detailed examination.

Fig. 15 shows a set of measurements taken during the design of a network which served as a combined loudspeaker equalizer and crossover network; the inductances employed had laminated iron cores. In this and the previous figure, the distortion shown includes that caused by the effect of the reactive load on the loudspeaker amplifier.

The stroboscope has been employed, amongst other things, in the examination of the various resonance modes of loudspeaker cones to investigate the connection between these modes and distortion levels. At higher frequencies the motion of electrically driven ribbon and moving-coil microphones has been studied at frequencies in excess of 10 kc/s.

Figs. 16 and 17 show photographs of the complete equipment.

5. Conclusion

The various published methods of measuring non-linear distortion have been shown to possess serious limitations.

A description is given of a new method by which the levels of the fundamental and the non-linear distortion products generated by equipment such as a loudspeaker may be measured as a continuous function of frequency. To obtain a complete picture of the transfer function,

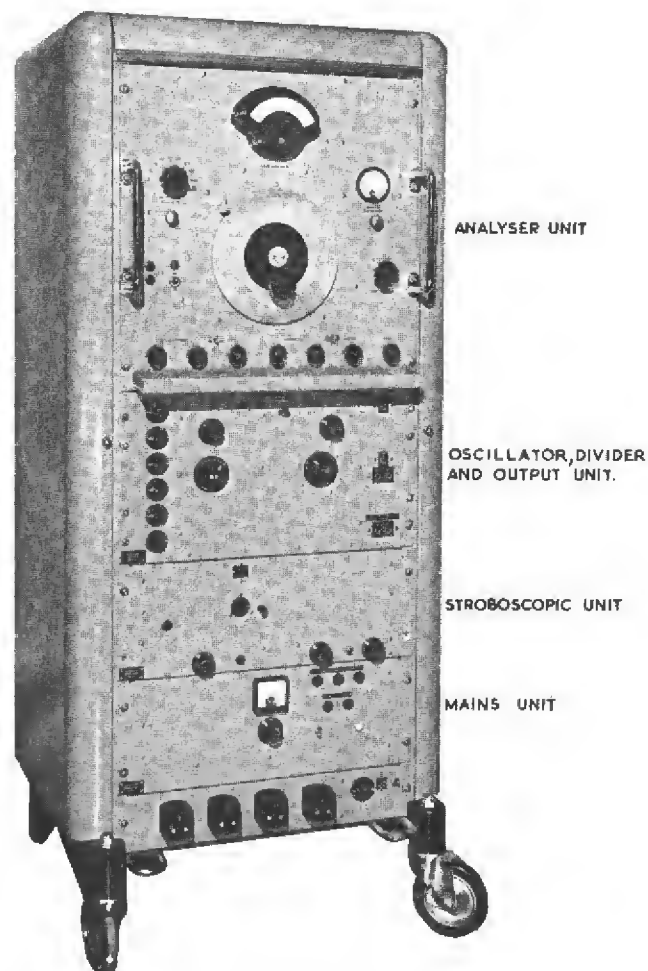


Fig. 16 — External view of apparatus.

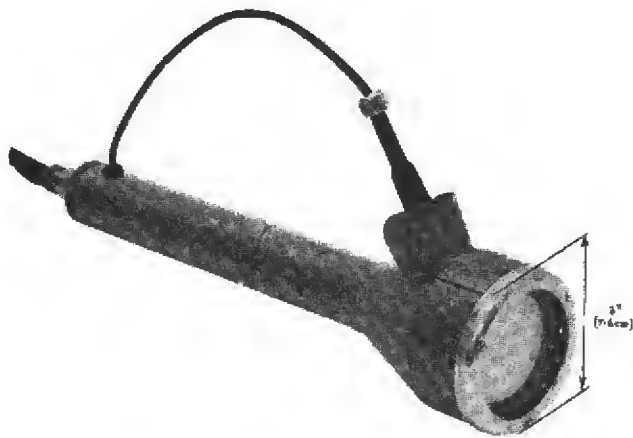


Fig. 17 — External view of light source.

measurements are made both of harmonic and of intermodulation distortion products. A method is also given which permits a stroboscope examination of the various loudspeaker resonance modes which may give rise to the non-linearity.

Equipment incorporating these ideas has been constructed and although using circuits which could be improved in the light of experience, has been adequate to indicate the potentialities of the method. For measurements of fundamental and harmonics the frequency range covered is 30 c/s to 16 kc/s and harmonics up to the 8th can be measured in this way. The accuracy and discrimination achieved are sufficient to permit measurements to be made of the smallest degree of non-linearity likely to be of interest in this field.

For intermodulation tests two signals 121 c/s apart are provided and are intended to be employed in the frequency range from about 500 c/s to 16 kc/s. Distortion products representative of even and odd powers in the transfer characteristic up to the eighth order can be measured,

again as a continuous function of frequency and down to a very low level.

Typical curves taken with the equipment are shown; measurements of this kind have proved of great value during the development of high-grade monitoring loudspeakers.

The stroboscope can be operated at repetition rates covering the whole audio-frequency band, and the illumination produced is independent of the repetition rate. A second signal is provided by which the object being examined can be driven at a frequency different, by 0 to 10 c/s, from that of the light source.

6. Acknowledgments

Messrs R. J. Packer and D. C. Derham constructed most of the equipment described and assisted in some of the design details.

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